IN THE CLAIMS

Please amend the claims as follows:

1. (Currently Amended) A method for controlling a Voice Over Internet Protocol (VoIP) call at a telephone endpoint, comprising:

tracking adaptation schemes used at the telephone endpoint for transmitting packets in the VoIP call;

monitoring a user response to the VoIP call that requests a different level of user perceived sound quality for the VoIP call; and

dynamically varying the adaptation schemes used at the telephone endpoint for transmitting the packets in the VoIP call from the telephone endpoint to correspond with the requested level of user perceived sound quality;

wherein dynamically varying the adaptation ~~parameters~~ schemes affects how much digital data is used to represent an audio signal.

2. (Previously Presented) A method according to claim 1 including;

initially transmitting the packets in the VoIP call over an Internet Protocol (IP) packet switched network using an IP packet best effort transmission scheme;

monitoring the user response for a request to increase sound quality; and

requesting reservation of IP packet switched network resources during the already established VoIP call when the increase sound quality request is detected from the user response prior to the reserved IP packet switched network resources being used during the VoIP call and without necessarily using the entire requested resources during the VoIP call.

3. (Previously Presented) A method according to claim 2 wherein requesting reservation of network resources comprises making an RSVP (Resource Reservation Protocol) request during the VoIP call.

4. (Original) A method according to claim 2 including conducting the already established VoIP call using reserved network resources when the requested reservation is accepted and the user response requests additional increases in the sound quality of the VoIP call.

5. (Original) A method according to claim 4 including increasing voice coder performance or reducing payload size after the network resources are reserved.

6. (Original) A method according to claim 1 including using a signal generated by an input device to detect the user response during the VoIP call.

7. (Original) A method according to claim 6 including using a dial or buttons on a telephone as the input device.

8. (Original) A method according to claim 6 including using a graphical user interface as the input device.

9. (Original) A method according to claim 1 including decoding Dual Tone Multiple Frequency signals to detect the user response.

10. (Currently Amended) A method for controlling a VoIP call, comprising:

tracking adaptation schemes used for transmitting packets in a Voice Over IP (VoIP) call;

monitoring a user response to the VoIP call;

dynamically varying the adaptation schemes used for transmitting the packets according to the monitored user response; and

monitoring congestion in a network used for conducting the VoIP call and varying the adaptation schemes according to the user response and the monitored congestion;

wherein dynamically varying the adaptation ~~parameters~~ schemes ~~affects a bandwidth of VoIP calls~~ includes either varying which coder algorithm is used at the telephone endpoint, varying a packet payload size of the packets or varying what type of Forward Error Correction (FEC) is used in association with the packets.

11. (Original) A method according to claim 1 wherein varying the adaptation schemes comprises varying codecs used for encoding audio signals into digital data making up the packets.

12. (Original) A method according to claim 1 including detecting a user response selecting a cost for the VoIP call and varying the adaptation schemes according to the selected cost.

13. (Currently Amended) An adaptation system, comprising:

an input for detecting a user response requesting a different user perceived sound quality for a call; and

a controller configured to dynamically vary adaptation parameters used for transmitting packets making up the call to correspond with the requested different user perceived sound quality in the user response detected by the input;

wherein dynamically varying the adaptation parameters affects how an analog signal is converted into the packets making up the call. ~~how much digital data is used to represent an audio signal~~.

14. (Previously Presented) An adaptation system according to claim 13 wherein the controller monitors congestion in the network carrying the call and selects which of the adaptation parameters to vary according to the monitored congestion and the requested user perceived sound quality.

15. (Original) An adaptation system according to claim 13 wherein the controller initially transmits the packets in the call using a best effort transmission scheme and during the call requests reservation of network resources when the user response requests increased sound quality.

16. (Previously Presented) An adaptation system according to claim 15 wherein the controller initiates an RSVP (Resource Reservation Protocol) request to reserve the network resources.

17. (Original) An adaptation system according to claim 15 wherein the controller monitors for acceptance of the network reservation request and modifies the adaptation parameters to provide an increased sound quality call when the acceptance is received.

18. (Original) An adaptation system according to claim 13 wherein the input comprises a dial or buttons.

19. (Original) An adaptation system according to claim 13 wherein the input comprises a graphical user interface.

20. (Original) An adaptation system according to claim 19 including a cost icon in the graphical user interface that allows selection of a call cost, the controller varying the adaptation parameters according to the selected call cost.

21. (Original) An adaptation system according to claim 13 wherein the input device generates Dual Tone Multiple Frequency signals that are decoded by the controller for identifying the user response.

22. (Original) An adaptation system according to claim 13 wherein the user response determines how much the controller varies the adaptation parameters.

23. (Original) An adaptation system according to claim 13 wherein the controller varies a rate that the packets are transmitted and received during the call.

24. (Previously Presented) An electronic storage medium containing software used for controlling a VoIP call, the software in the electronic storage medium comprising:

code for tracking adaptation schemes used for transmitting audio packets in a Voice Over IP (VoIP) call;

code for monitoring a user response to the VoIP call indicating a desired level of user perceived audio quality for the VoIP call; and

code for dynamically varying the adaptation schemes used for transmitting the audio packets from a telephone endpoint so that the user perceived audio quality of the VoIP call corresponds with the monitored user response.

25. (Original) An electronic storage medium according to claim 24 including;

code for initially transmitting the packets in the VoIP call using a best effort transmission scheme;

code for monitoring the user response for a request to increase voice quality; and

code for requesting reservation of network resources during the already established VoIP call when the increase voice quality request is detected from the user response.

26. (Previously Presented) An electronic storage medium according to claim 25 including code that requests reservation of network resources by making an RSVP (Resource Reservation Protocol) request in the middle of the VoIP call.

27. (Original) An electronic storage medium according to claim 25 including code for conducting the already established VoIP call using reserved network resources when the requested reservation is accepted and the user response requests additional increases in voice quality of the VoIP call.

28. (Original) An electronic storage medium according to claim 27 including code for increasing voice coder quality and reducing packet payload size for the packets in the VoIP call after the network resources are reserved.

29. (Original) An electronic storage medium according to claim 24 including code that detects the user response from a signal generated by an input device controllable by a user during the VoIP call.

30. (Original) An electronic storage medium according to claim 29 wherein the input device comprises a dial on a telephone.

31. (Original) An electronic storage medium according to claim 29 wherein the input device comprises a graphical user interface on a computer.

32. (Original) An electronic storage medium according to claim 24 including code that decodes Dual Tone Multiple Frequency signals to identify the user response.

33. (Original) An electronic storage medium according to claim 24 including code for monitoring congestion in a network used for conducting the VoIP call and varying the adaptation schemes according to the user response and the monitored congestion.

34. (Original) An electronic storage medium according to claim 24 including:

code for varying codecs used for encoding audio signals into digital data making up the audio packets;

code for varying a rate that the audio packets are transmitted and received during the VoIP call;

code for varying an amount of audio data in the audio packets; and

code for adding or removing error correction information from the audio packets.

35. (Original) An electronic storage medium according to claim 24 including code for detecting a user response selecting a cost for the VoIP call and varying the adaptation schemes according to the selected cost.

36. (Previously Presented) A system for controlling a VoIP call, comprising:

means for tracking adaptation schemes used for transmitting audio packets in a Voice Over IP (VoIP) call;

means for monitoring a user response to the VoIP call indicating a desired level of user perceived audio quality for the VoIP call; and

means for dynamically varying the adaptation schemes used for transmitting the audio packets from a telephone endpoint so that the user perceived audio quality of the VoIP call corresponds with the monitored user response.

37. (Currently Amended) A system for controlling a VoIP call, comprising:

means for tracking adaptation schemes used for transmitting audio packets in a Voice Over IP (VoIP) call;

means for monitoring a user response to the VoIP call;

means for dynamically varying the adaptation schemes used for transmitting the audio packets according to the monitored user response, dynamically varying the adaptation schemes changing how an analog audio signal is packetized into the audio packets;

means for initially transmitting the packets in the VoIP call using a best effort transmission scheme;

means for monitoring the user response for a request to increase voice quality; and

means for requesting reservation of network resources for the call during the already established VoIP call when the increase voice quality request is detected from the user response.

38. (Previously Presented) A system according to claim 37 including means for requesting reservation of network resources by making an RSVP (Resource Reservation Protocol) request in the middle of the VoIP call.

39. (Original) A system according to claim 37 including means for conducting the already established VoIP call using reserved network resources when the requested reservation is accepted and the user response requests additional increases in voice quality of the VoIP call.

40. (Original) A system according to claim 38 including means for increasing voice coder quality and reducing packet payload size for the packets in the VoIP call after the network resources are reserved.

41. (Original) A system according to claim 36 including means for detecting the user response from a signal generated by an input device controllable by the user during the VoIP call.

42. (Original) A system according to claim 36 including means for detecting the user response from a dial on a telephone.

43. (Original) A system according to claim 36 including means for detecting the user response from a graphical user interface on a computer.

44. (Original) A system according to claim 36 including means for decoding Dual Tone Multiple Frequency signals to monitor the user response.

45. (Original) A system according to claim 36 including means for monitoring congestion in the network used for conducting the VoIP call and varying the adaptation schemes having a best chance with the monitored congestion of adapting the VoIP call to the user response.

46. (Original) A system according to claim 36 including:

means for varying codecs used for encoding audio signals into digital data making up the audio packets;

means for varying a rate that the audio packets are transmitted and received during the VoIP call;

means for varying an amount of audio data in the audio packets; and

means for adding or removing error correction information from the audio packets.

47. (Currently Amended) A system according to claim ~~24~~ 36 including means for detecting a user response selecting a cost for the VoIP call and means for varying the adaptation schemes according to the selected cost.

48. (Currently Amended) A method for controlling a call, comprising;

establishing a call over a Plain Old Telephone System (POTS);

~~converting the call to a packetized call~~;

generating Dual Tone Multiple Frequency (DTMF) signals to request a controller to modify a sound quality of the ~~packetized~~ call; ~~and~~

packetizing the call at a network device that is coupled to a packet switched network;

~~detecting the DTMF signals and~~ modifying adaptation parameters in response to the DTMF signals to modify the sound quality of the packetized call.

49. (Previously Presented) A method for controlling a Voice Over Internet Protocol (VoIP) call, comprising:

tracking adaptation schemes used for transmitting packets in the VoIP call;

monitoring a user response to the VoIP call that requests a different level of user perceived sound quality for the VoIP call; and

dynamically varying the adaptation schemes used for transmitting the packets in the VoIP call to correspond with the requested level of user perceived sound quality;

wherein dynamically varying the adaptation schemes includes adjusting Forward Error Correction (FEC) and adjusting packet payload length.

50. (New) The method of claim 1 further comprising:

listening to an audible signal after dynamically varying the adaptation scheme to determine a level of user perceived sound quality for the audible signal;

further dynamically varying the adaptation schemes to improve the audible signal when the user perceived quality of the audible signal is low; and

further listening to the improved audible signal to determine a level of user perceived sound quality for the improved audible signal.